APPARATUS AND METHOD FOR AUDIO-SIGNAL-PROCESSING

BACKGROUND OF THE INVENTION

[0001] 1. Field of the Invention

[0002] This invention relates to an audio-signal-processing apparatus capable of harmonic-series-generation and a method thereof.

[0003] 2. Description of the Related Art

[0004] Audio-signal-processing apparatuses with harmonic-series-generation have been proposed for various purposes. Generally known is, for example, an effector apparatus that gives variations to tone color of a musical instrument and voice, especially, a virtual reproduction system of bass sound that generates a harmonic-tone for a bass component to reinforce the bass sounds in a small speaker.

[0005] In the followings, a virtual reproduction system of bass sound in a prior art will be described, referring to from Fig. 9 through Fig. 13. Figs. 9 (a) and (b) are block diagrams illustrating the conventional audio-signal-processing apparatus. Each harmonic-tone can be generated with a full-wave-rectification method, a power method, a zerocrossing method and so on.

[0006] First, a first example in the prior art will be described using Fig. 9 (a). In this example, a plurality of harmonic-tone components of integer orders are generated with the zerocrossing method or the power method for low frequency components.

[0007] As shown in Fig. 9 (a), a signal inputted from an input terminal 91 is divided into two systems. In a first system, all the components of the input signal are inputted into a delay unit 93, and it is inputted into one input unit of an adder 100 after undergoing gain adjustment by a gain-control 94a, delayed by a time that is required for processing in a harmonic-series-generating unit 97, whose processing will be described later.

[0008] In a second system, all the components of the input signal are inputted into a low pass filter 96. The low pass filter 96 extracts only low frequency components from all

these components, according to a predetermined cut-off characteristic of the low pass filter 96, and outputs them to harmonic-series-generating unit 97.

[0009] In this specification, a harmonic-tone with n times the frequency (n is a natural number) of the fundamental tone frequency (fundamental frequency) is called an n-th harmonic-tone.

[0010] The harmonic-series-generating unit 97 comprises harmonic-series-generating units 98a, 98b, ..., 98c that generate a 2nd harmonic-tone, a 3rd harmonic-tone, ..., an n-th harmonic-tone for the low frequency components that the low pass filter 96 has extracted. The 2nd harmonic-tone and the 3rd harmonic-tone, ..., the n-th harmonic-tone that these harmonic-series-generating units 98a, 98b, ..., 98c generate are inputted into the other input unit of the adder 100 after undergoing gain adjustment by gain controls 94b, 94c, ..., 94d respectively and addition in an adder 95.

[0011] The adder 100 adds the components inputted from the first system and the second system respectively and outputs the sum to an output terminal 92.

[0012] Next, a second example in the prior art will be described using Fig. 9 (b). In this example, a harmonic-tone is generated for the low frequency component with the full-wave-rectification method. Published Japanese Patent Lai-Open No. 05-328481 disclosed an art pertaining to the second example.

[0013] In Fig. 9 (b), the explanation is omitted by attaching the same numbers and marks as for the same components with Fig. 9 (a).

[0014] In Fig. 9 (b), the harmonic-series-generating unit 97 in the Fig. 9 (a) is replaced with the following component.

[0015] A full wave rectifier 99 changes negative values of the low frequency signals that a low pass filter 96 has extracted to positive values, doubling the frequency of the low frequency signals. However, since direct-current bias and even harmonic-tone components are actually generated, only the 2nd harmonic-tone that is a principal component, is extracted using a band pass filter 101.

[0016] As shown in Fig. 9 (b), the full-wave-rectification method can be executed with simple structure. The full-wave-rectification method can generate a second harmonic-tone and can be expanded so as to generate harmonic-series of the order of an n-th power of 2 (n = 2, 3, ...) by cascading a plural units of the full-wave-rectification 99 and the band pass filter 101 as shown in Fig 9 (c). However, the full-wave-rectification method has a problem that odd harmonic-tone cannot be generated.

[0017] Next, some problems in the zerocrossing method and the first example in the prior art will be described using Fig. 10. Figs. 10 (a) and (b) show graphs exemplifying waveforms in the prior art. In the figures, the horizontal axis shows time and the vertical axis shows amplitude.

[0018] The zerocrossing method detects the zerocrossing points P1, P2, and P3 that a signal changes from positive to negative or from negative to positive as shown in circles in Fig. 10 (a).

[0019] In generating the second harmonic-tone component, a signal is double-compressed in the time axis and repeatedly reproduced twice in an interval between one zerocrossing point and the next zerocrossing point (for example, an interval between P1 and P2 or an interval between P2 and P3, in Fig. 10 (a)).

[0020] Similarly, an n-th order harmonic-tone is obtained by compressing a signal n times and repeatedly reproducing the signal n times in an interval between two consecutive zerocrossing points on the time axis.

[0021] A general music source is a complex tone that is comprised of a plurality of pure tone components. For example, a periodical musical sound is comprised of a fundamental tone with (f= 40Hz in Fig. 11 (a)) and harmonic-tones with frequencies of integral multiples of the fundamental frequency (80, 120 ... Hz in Fig. 11 (a)), as shown in spectrum structure examples in Fig. 11 (a).

[0022] Further, a chord possesses a plurality of fundamental frequencies with strong energy components. For example, the chord possesses spectrum structure shown in Fig.

11 (b) in Perfect 8th, and spectrum structure shown in Fig. 11 (c) in Perfect 5th.

[0023] Next, problems encountered when the conventional harmonic-series-generating methods are used for such general complex tones as are comprised of a plurality of pure tone will be described. Hereafter a case when the zerocrossing method is used as a harmonic-series-generating method will be described. The following problems will also be encountered when other harmonic-series-generating methods such as the full-wave-rectification method and the power method are used.

[0024] Fig. 12 (a) shows the results of the frequency analysis performed on the processed sounds, after generating the 2nd harmonic-tone at the same level as the original signal, by the zerocrossing method, for the complex tone (refer to Fig. 11 (b) regarding its spectrum structure) that are comprised of two pure tones (pure tones of 40Hz and 80Hz of the same level) that are in the Perfect 8th relation (the frequency ratio is 1:2).

[0025] Fig. 12 (b) shows the results of the frequency analysis performed on the processed sounds, after generating the 2nd harmonic-tone at the same level as the original signal, by the zerocrossing method, for the complex tone (refer to Fig. 11 (c) regarding its spectrum structure) that are comprised of two pure tones (pure tone of 60Hz and 90Hz of the same level) that are in the Perfect 5th relation (the frequency ratio is 2:3).

[0026] It is desirable that the processed sounds should possess only the second harmonic components of each pure tone component in both Figs. 12 (a) and (b). However, although the second harmonic components are generated in either case, some distortions have occurred due to the inclusion of other components than the second harmonic components (the other components consisting of harmonic tones that have the fundamental frequency equal to the highest common factor of the fundamental frequency of two pure tone components).

[0027] This distortion occurs since the harmonic-tone is generated at once for the signal

with a plurality of pure tones.

[0028] That is, as shown in Fig. 13 (a), in a signal with a plurality of pure tones, the waveform cycle T of the complex tone (the least common multiple of the contained pure tone cycles) and the interval of the zerocrossing points in the original waveform do not usually agree with each other.

[0029] For such a case, applying the zerocrossing method compresses the waveform and makes reproduction repeatedly in other interval than the zerocrossing points with cycle T, and as shown in Fig. 13 (c). As a result, the processed signal waveform is not similar to the original signal and the waveform distortion occurs. Make sure to compare carefully Fig. 13 (b) with Fig. 13 (c), where Fig. 13 (b) shows the waveform that has the ideal harmonic-series-generation of the original waveform of Fig. 13 (a).

OBJECTS AND SUMMARY OF THE INVENTION

[0030] In view of the above, an object of the present invention is to provide an audio-signal-processing apparatus that can suppress sound quality degradation in a complex tone and a method thereof.

[0031] A first aspect of the present invention provides an audio-signal-processing apparatus comprising: a band-decomposition unit, having a decomposition characteristic, operable to decompose a low frequency component of input-audio-signals into a plurality of frequency components that have different frequency bands based on the decomposition characteristic; a harmonic-series-generating unit operable to generate a harmonic-tone component based on at least one of the plurality of frequency components; and a composition unit operable to compound the input-audio-signals and the harmonic-tone component generated by the harmonic-series-generating unit.

[0032] With this structure, since the band-decomposition unit decomposes the low frequency component into a plurality of narrow frequency bandwidths, a situation that the fundamental tone and the harmonic-tones thereof belong to a same band is suppressed, thereby the distortion can be restrained and sound quality can be improved.

[0033] A second aspect of the present invention provides an audio-signal-processing apparatus as defined in the first aspect of the present invention, wherein the band-decomposition unit is operable to decompose the low frequency component of the input-audio-signals such that each of a fundamental tone and harmonic-tones of the fundamental tone.

[0034] With this structure, the fundamental tone and the harmonic-tones thereof do not belong to a same band, thereby the distortion can be restrained and sound quality can be improved.

[0035] A third aspect of the present invention provides an audio-signal-processing apparatus as defined in the first aspect of the present invention, wherein the decomposition characteristic is defined based on a lowest fundamental frequency of musical instruments.

[0036] With this structure, the band-decomposition unit decomposes the low frequency component according to actual circumstances.

[0037] A fourth aspect of the present invention provides an audio-signal-processing apparatus as defined in the first aspect of the present invention, wherein the decomposition characteristic is defined based on a low interval limit.

[0038] With this structure, the decomposition characteristic for band-decomposition can be optimized, and the circuit scale can be reduced.

[0039] A fifth aspect of the present invention provides an audio-signal-processing apparatus as defined in the first aspect of the present invention, wherein each band width of the different frequency bands is from 15Hz to 30Hz.

[0040] With this structure, the decomposition characteristic is adapted to the low interval limit.

[0041] A sixth aspect of the present invention provides an audio-signal-processing apparatus as defined in the first aspect of the present invention, further comprising a delay device operable to compensate processing delay between the harmonic-tone

component and the input-audio-signals.

[0042] With this structure, the generated harmonic-tone components and the input-audio-signals can be adjusted on the time axis of the reproduced sound.

A seventh aspect of the present invention provides an audio-signal-processing apparatus as defined in the first aspect of the present invention, further comprising a gain control device operable to adjust a gain of the input-audio-signals and a gain of the harmonic-tone component generated by the harmonic-series-generating unit.

[0043] With this structure, a level ratio of the generated harmonic-tone component and the input-audio-signals can be changed; thereby a plurality of reinforcement effect modes how the bass sounds can be set.

An eighth aspect of the present invention provides an audio-signal-processing comprising: a sum component output unit operable to input input-audio-signals of a first channel and input-audio-signals of a second channel to output a sum component of the audio signals of the first channel and the input-audio-signals of the second channel; a band-decomposition unit, having a decomposition characteristic, operable to decompose the sum component into a plurality of frequency components that have different frequency bands based on the decomposition characteristic; a harmonic-series-generating unit operable to generate a harmonic-tone component based on at least one of the plurality of frequency components; a first composition unit operable to compound the input-audio-signals of the first channel and the harmonic-tone component generated by the harmonic-series-generating unit; and a second composition unit operable to compound the input-audio-signals of the second channel and the harmonic-tone component generated by said harmonic-series-generating unit.

[0044] This structure earns the same advantage as the first invention. In addition, utilizing human audibility characteristic that is dull at localization of sound image in low frequencies, and the fact that low frequency components are often contained in-phase in each channel, reinforcement of the bass sounds can be realized on a smaller

circuit scale when processing the input-audio-signals of two or more channels.

The above, and other objects, features and advantages of the present invention will become apparent from the following description read in conjunction with the accompanying drawings, in which like reference numerals designate the same elements.

BRIEF DESCRIPTION OF THE DRAWINGS

[0045] Figs. 1 (a), (b), and (c) are block diagrams illustrating an audio-signal-processing apparatus according to the embodiment of the present invention;

[0046] Fig. 2 is a block diagram illustrating a band-composition unit according to the embodiment of the present invention;

[0047] Fig. 3 is a descriptive illustration showing a band-decomposition according to the embodiment of the present invention;

[0048] Fig. 4 is a descriptive illustration showing a frequency interval that takes into consideration a low interval limit according to the embodiment of the present invention;

[0049] Figs. 5 (a) - (d) are exemplified illustrations showing a band-decomposition according to the embodiment of the present invention;

[0050] Figs. 6 (a) - (d) are exemplified illustrations showing a band-decomposition according to the embodiment of the present invention;

[0051] Figs. 7 (a) - (d) are exemplified illustrations showing a band-decomposition according to the embodiment of the present invention;

[0052] Fig. 8 is a block diagram showing an audio-signal-processing apparatus that corresponds to two channels according to the embodiment of the present invention;

[0053] Figs. 9 (a) and (b) are block diagrams illustrating a conventional audio-signal-processing apparatus;

[0054] Fig. 10 (a) is a descriptive illustration showing how a prior art zerocrossing method generates a harmonic-tone (original waveform);

[0055] Fig. 10 (b) is a descriptive illustration showing how the prior art zerocrossing method generates the harmonic-tone (generated waveform);

[0056] Figs. 11 (a) - (c) are descriptive illustrations showing a chord spectrum structure;

[0057] Figs. 12 (a) and (b) are illustrations showing a spectrum of the result of the prior art of the harmonic-series-generation (complex tone);

[0058] Fig. 13 (a) is a descriptive illustration showing how the prior art zerocrossing method generates the harmonic-tone (original waveform);

[0059] Fig. 13 (b) is a descriptive illustration showing how the prior art zerocrossing method generates the harmonic-tone (generated waveform); and

[0060] Fig. 13 (c) is a descriptive illustration showing how the prior art zerocrossing method generates the harmonic-tone (ideally generated waveform).

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0061] Preferred embodiments of the present invention are now described in conjunction with the accompanying drawings. Fig. 1 (a) is a block diagram illustrating an audio-signal-processing apparatus according to the present embodiment.

An input terminal 11 shown in Fig. 1 inputs an input-audio-signal. A band-decomposition unit 13 has a decomposition characteristic, and it decomposes the low frequency components of the input-audio-signal into a plurality of frequency components that have different frequency bands based on the decomposition characteristic.

[0062] A plurality of harmonic-series-generating units 14a-14c generate harmonic-tone components for each band signal decomposed by the band-decomposition unit 13. As a harmonic-series-generating method, any harmonic-tone-component method that are described in the prior (the full-wave-rectification method, the power method, the zerocrossing method, etc.) can be used.

[0063] However, the numbers and amplitude levels of the harmonic-tone components in each frequency band are set to desirable values based on a listening test and so on.

[0064] A high-pass filter 15 is a digital filter with a high-pass characteristic and removes the components within the frequency bands to be virtualized from the

input-audio-signal.

[0065] A delay device 16 compensates the processing delay. Gain controls 17a-17d adjust the gain for the output signals of harmonic-series-generating units 14a-14c and the delay device 16. An adder 18 adds the output signals of the gain control 17a-17d, and the added result is outputted to the exterior from an output terminal 12.

[0066] The band-composition unit 13 decomposes the low frequency components into a plurality of frequency band signals (hereafter referred as the band signal). In the digital-signal-processing art, methods to generate the band signals are generally known; a band pass filter, Fourier transformation, and wave let conversion, etc. may be used.

[0067] The band-decomposition unit 13 in the present invention is enough to decompose the bands following the condition 1 described later and it does not depend on the composition itself. As shown in Fig. 2, setting a plurality of band pass filters 23a, 23b... 23c in parallel according to the bandwidth comprises the band-decomposition unit 13.

[0068] The decomposition characteristic of the band-decomposition unit 13 is comprised so that a plurality of pure tone components are not contained in each frequency band to suppress the sound quality deterioration in complex tones. The details how to determine such a decomposition characteristic will be described later.

[0069] In place of the structure shown in Fig.1 (a), the components of the frequency band to be virtualized from the original signal can be removed by the high pass filter 15 from the output of the adder 18 as shown in Fig.1 (b).

[0070] In place of the structure shown in Fig.1 (a), there is no practical problem to omit the high pass filter 15 as shown in Fig.1 (c), though the original frequency components still remain.

[0071] Next, the operations will be described. In the followings, the decomposition characteristic shown in Fig. 3 is simplified such as [f0Hz f1Hz f2Hz f3Hz]. This expression means that there are three bands such as an f0 Hz-f1Hz band, an f1 Hz-f2Hz

band, and an f2 Hz-f3Hz band.

[0072] In the following, a case is described where the total bandwidths of the three bands, that is, the frequency band of [f0Hz f3Hz] as shown in Fig. 3, are to be generated. In the following example, it is assumed that f0=0, f1=50, f2=75, and f3=100.

[0073] First, as shown in Fig. 1 (a), the input-audio-signal is inputted into the band-decomposition unit 13 through the input terminal 11.

[0074] The band-decomposition unit 13 extracts the low frequency component from the input-audio-signal and decomposes its component into the frequency band signals (hereafter referred to the band signal) of [0Hz 50Hz], [50Hz 75Hz], [75Hz 100Hz].

[0075] The harmonic-tone components are generated for the each band signal by the harmonic-series-generating units 14a-14c.

[0076] The input-audio-signal that is to be recomposed with the output signals of the harmonic-series-generating units 14a-14c, is attenuated in the frequency band components of [f0Hz f3Hz] by the high pass filter 15, and delayed by the delay device 16, in order to compensate the delay produced by the harmonic-series-generating process of the low frequency component.

[0077] The gains of the output signal of the harmonic-series-generating units 14a-14c and the output signal of the delay device 16 are adjusted adequately by the gain adjustment units 17a-17d. The adjusted signals are added by the adder 18 and are outputted to the exterior through the output terminal 12.

[0078] Decomposing the whole bands into a plurality of narrow bands makes it feasible to generate the harmonic-tone components individually for each pure tone component that comprises the complex tone, thus suppressing the sound quality deterioration in the complex tone.

[0079] However, if the decomposing number of the bands increases, the harmonic-series-generating units as many as the decomposing number are needed, and narrower band filters is also required, resulting in increased circuit scale.

[0080] Therefore, considering the reduction of the circuit scale while maintaining sound quality, it is an optimal decomposition characteristic that has the widest band width to be able to separate each purer tone component.

[0081] The followings will describe that the each decomposing bandwidth should be determined based on the harmonic-tone composition of the musical sound and the low interval limit.

[0082] 1. Harmonic-tone Composition of Musical Sound

[0083] Periodical complex tones like musical sound are comprised of a fundamental tone with fundamental frequency and harmonic-tones with the frequencies of integral multiple.

[0084] The lowest fundamental frequency differs with musical instruments. The following shows the lowest fundamental frequency of the main bass musical instruments (hereafter referred to the lowest fundamental frequency).

Musical instruments[0085] [0085] The lowest fundamental frequency (Hz)

Piano[0086] [0086] [0086] [0086] 27.5

Bass guitar (five bowstrings)[0087] 30.8

Contrabass[0088] [0088] [0088] 34.6

Bus tuba[0089] [0089] [0089] 41.2

Bus saxophone[0090] [0090] [0090] 55.0

[0091] According to the above list, there is no problem to say that 27.5Hz, the lowest fundamental frequency of a piano is the lowest in most bass musical instruments. Therefore, it is enough for the decomposition characteristic to correspond to a fundamental frequency equal to or higher than 27.5Hz that is the lowest fundamental frequency of a piano.

[0092] Since the harmonic-tone components contained in musical sound exist generally in the higher degrees like the 2nd, the 3rd, and the 4th ..., it is desirable that the fundamental tone and the each harmonic-tone component are separated respectively.

[0093] Considering the above, it is enough if the following Condition 1 is fulfilled to each fundamental frequency of 27.5Hz or higher.

[0094] (Condition 1) The fundamental tone with a fundamental frequency (frequency f) and a harmonic-tone for the fundamental tone (frequency nf; n being a natural number) are not contained in the same decomposing band (i.e., the fundamental tone with a fundamental frequency and a harmonic-tone for the fundamental tone belong to different frequency bands).

[0095] 2. Setting based on Low Interval Limit

[0096] A chord has a plurality of fundamental frequencies. Therefore, in order to separate each pure tone component in the chord, it is required to separate fundamental frequency of each fundamental tone and its harmonic-tone components.

[0097] In a lower register, there exists a characteristic on the audibility that the chord with the close fundamental frequencies produces muddy sounds without harmonizing.

[0098] There is a limit that if two sounds with a determined interval are produced in a register lower than this, it will sound muddy. This limit is called a low interval limit. Because of the low interval limit, the fundamental frequencies that produce simultaneously are restricted.

[0099] Fig. 4 illustrates calculated frequency intervals of the fundamental frequencies based on the low interval limit. The relationships among the fundamental frequency, the musical scale and the frequency interval are summarized in Fig. 4.

[0100] Fig. 4 shows that the interval of only Perfect eighth in A0-A1, Perfect fifth and Perfect eighth in A1#-A2, the interval higher than Major 3rd except Augmented 4th in A2#-B2, the interval higher than Minor 3rd in C2-D2, and the interval higher than Major 2nd in higher than D2# are used.

[0101] The suffix attached to the interval name expresses the octave number, and A0 corresponds with 27.5Hz that is the lowest fundamental tone of a piano.

[0102] If the Condition 1 is fulfilled regarding the fundamental frequency higher than

29Hz, the conditions of the low interval limit can also be fulfilled.

[0103] For example, Perfect fifth (the frequency ratio is 2:3) is used from 58Hz. The 58Hz is a 2nd harmonic-tone of 29Hz and the frequency of 87Hz of a 3rd harmonic-tone has the frequency ratio of Perfect fifth for the 2nd harmonic-tone.

[0104] That is, Perfect fifth is equal to the frequency ratio of the 2nd harmonic-tone and the 3rd harmonic-tone, and the lowest frequency 58Hz that Perfect fifth is used corresponds with the 2nd harmonic-tone of 29Hz. Therefore, if the fundamental frequencies higher than 29Hz fulfill the Condition 1, the fundamental frequency that is in the relation of Perfect fifth higher than 58Hz is separable.

[0105] This also corresponds to Perfect eighth (the frequency ratio of the fundamental frequency and the 2nd harmonic-tone) and Major 3rd (the frequency ratio of the 4th harmonic-tone and the 5th harmonic-tone). After all, if the Condition 1 is fulfilled regarding the fundamental frequency higher than 29Hz, the fundamental frequency that interferes with bass chords is also separable.

[0106] Next, the practical examples of the decomposition characteristic that fulfill the Condition 1 will be described. The lowest fundamental frequency here is 27.5Hz that is the lowest fundamental frequency of a piano.

[0107] As a simple example fulfilling the Condition1, it is considered to make each bandwidth narrower than 27.5Hz uniformly. However, since ideal band-decomposition without a crossover is impossible, the bandwidth narrower than 27.5Hz will be practically set.

[0108] As examples of such decomposition, the bandwidths of [25Hz 50Hz 75Hz 100Hz 125Hz] and [12.5Hz 37.5Hz 62.5Hz 112.5Hz] are illustrated in Figs. 5 (a) and (b) respectively.

[0109] As shown in the bandwidths of [25Hz 50Hz 75Hz 100Hz 107.5Hz] [12.5Hz 37.5Hz 62.5Hz 87.5Hz 107.5Hz] in Figs. 5 (c) and (d) respectively, each bandwidth needs not necessarily be equal.

[0110] Since the fundamental frequencies of the bandwidth of [0Hz 27.5Hz] exist few, it is not necessary to take this bandwidth into consideration.

[0111] Therefore, as shown in the bandwidth of [0Hz 50Hz 75Hz 100Hz 125Hz] [0Hz 37.5Hz 62.5Hz 87.5Hz 112.5Hz] in Figs. 6 (a) and (b) respectively, the lowest band can be a low pass filter.

[0112] Setting the bandwidth wider than 27.5Hz is also conceivable. For example, as shown in Fig. 6 (c), a part of the bandwidth can be set to 35Hz.

[0113] This decomposition characteristic also fulfills the Condition 1. There are other such examples.

[0114] Regarding the decomposition characteristic higher than 150Hz, if the separation of each Minor 3rd and Major 2nd is possible, sound quality will be improved further.

[0115] In an example of [25Hz 50Hz 75Hz 100Hz 125Hz 145Hz 165Hz] shown in Fig. 6 (d), Minor 3rd is separable, the bandwidth higher than 125Hz is set to 20Hz.

[0116] It is also possible to make bandwidth narrower than 20Hz in the above example. For example, as shown in Figs. 7 (a) and (b), it is also possible to set the bandwidth to 15Hz. In this case, similarly as in the case described above, the bandwidths do not need to be equal (Fig. 7 (c)) and a low pass filter may be used in the lowest region (Fig. 7 (d)).

[0117] As mentioned above, the sound quality deterioration can be suppressed by generating the harmonic-tone components individually for each pure tone component that comprises the complex tone through the band-decomposition process. Setting the decomposition characteristic based on the harmonic-tone composition of musical sound and the low interval limit under the Condition 1 optimizes the decomposition characteristic of the band-decomposition and reduces the circuit scale.

[0118] In the explanation above, the lowest fundamental frequency under the Condition 1 has been set to 27.5Hz that is the fundamental frequency of a piano. Actually, if the lowest fundamental frequency is set to 30Hz, the setting adapts to the low interval limit.

However, if the lowest fundamental frequency is set to the higher frequency up to 50Hz in order to reduce cost, such setting is also included in the present invention and has practically sufficient effect.

[0119] Considering the above fact, it is desirable that the bandwidth in the decomposition characteristic of the band-decomposition unit 13 is in the range from 15Hz to 50Hz.

[0120] The above description explains for the case that the input-audio-signal is monaural; however, the present invention is also applicable to sources of two or more channels.

[0121] Fig. 8 is a block diagram illustrating an audio-signal-processing apparatus corresponding to two channels according to an embodiment of the present invention. In the figure, the explanation is omitted by attaching the same numbers and marks as for the same components with Fig. 1.

[0122] Since this apparatus corresponds to two channels, the 1st channel (L channel in the figure) and the 2nd channel (R channel in the figure), input terminals 11a and 11b, delay devices 16a and 16b, gain controls 17a and 17e, adders 18a and 18b, and high-pass filters 15a and 15b are provided to each channel respectively.

[0123] An adder 81 as a sum component output unit inputs the input-audio-signal of the 1st channel and the input-audio-signal of the 2nd channel, and outputs the sum component of these input-audio-signals to a band-decomposition unit 13. Therefore, this band-decomposition unit 13 has a decomposition characteristic, and decomposes the low frequency components of the sum component into a plurality of frequency components that have different frequency bands according to the decomposition characteristic.

[0124] The adder 18a as a 1st composition unit adds the harmonic-tone components generated by harmonic-series-generating units 14a-14c and the input-audio-signal of the 1st channel, and outputs them to the high-pass filter 15a.

[0125] The adder 18b as a 2nd composition unit adds the harmonic-tone component generated by the harmonic-series-generating units 14a-14c and the input-audio-signal of the 2nd channel, and outputs them to the high-pass filter 15b.

[0126] The apparatus shown in Fig. 8 reduces the circuit scale by utilizing the fact that the low frequency components are contained in-phase in each channel in many cases and by processing the each channel signal after adding. The examples of two channels (stereo) have been described; however, the present invention is applied to three channels or more, for example, 5.1 channels and so on.

[0127] As an example, the audio-signal-processing apparatus of the present invention is applied to the virtual reproduction system of bass sound that generates the harmonic-tone in the low frequency component. However, the present invention is applicable to various uses such as an effector apparatus and so on, as long as it is an audio-signal-processing apparatus that generates the harmonic-tone.

[0128] According to the present invention, a band-decomposition process can generate a harmonic-tone individually to each pure tone component that comprises complex tone, reduces distortion, and suppress the sound quality deterioration.

[0129] Comprising a decomposition characteristic based on the harmonic-tone composition of musical sound and the low interval limit can optimizes the decomposition characteristic of the band-decomposition and can reduce the circuit scale.

[0130] Having described preferred embodiments of the invention with reference to the accompanying drawings, it is to be understood that the invention is not limited to those precise embodiments, and that various changes and modifications may be effected therein by one skilled in the art without departing from the scope or spirit of the invention as defined in the appended claims.